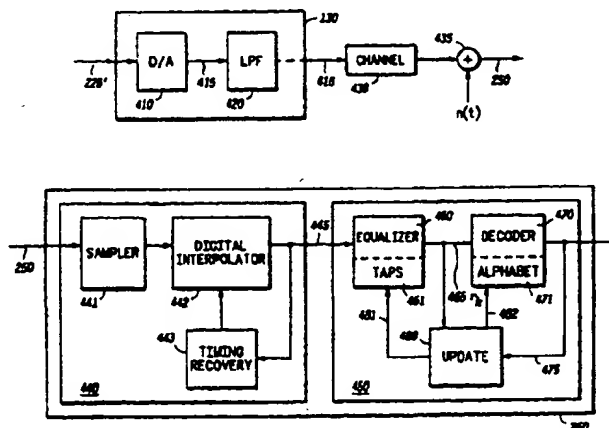




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(71) Applicant: MOTOROLA INC. [US/US]; 1303 East Algonquin Road, Schaumburg, IL 60196 (US).			
(72) Inventors: EYUBOGLU, M., Vedat; 150 Jennie Dugan Road, Concord, MA 01742 (US). HUMBLET, Pierre, A.; 65, boulevard Montfleury, F-06400 Cannes (FR).			
(74) Agents: WOOD, J., Ray./JRW et al.; Motorola Inc., Intellectual Property Dept., 1303 East Algonquin Road, Schaumburg, IL 60196 (US).		Published With international search report.	

(54) Title: SYSTEM AND DEVICE FOR, AND METHOD OF, PROCESSING BASEBAND SIGNALS TO COMBAT ISI AND NON-LINEARITIES IN A COMMUNICATION SYSTEM



## (57) Abstract

A system and device for, and method of, processing baseband signals to combat ISI and non-linearities on a communication system having a local loop. An actual alphabet is formed (471) from the signals actually transmitted on a channel (250). The alphabet may be used for symbol decoding (470), for example, and may avoid erroneous symbol predictions that could occur if an ideal or proscribed alphabet were used. Conventional phone systems have local loops with conventional line interfaces believed to have non-linearities vis-a-vis the proscribed companding algorithm. In particular, baseband signals created from the inverse quantization mechanisms inherent in conventional line interfaces have non-linear distortions. The actual alphabet therefore corresponds to the low fidelity signals actually transmitted by the line interface (250), with each symbol being an estimate of the signals actually transmitted (475). The estimate may be formed from an averaging function. An equalizer (460) may be used to combat ISI and other channel distortion. The alphabet and the equalizer may be updated with an error signal that is indicative of the accuracy of the alphabet's estimates (481).

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**System and Device for, and Method of, Processing  
Baseband Signals to Combat ISI and Non-linearities in  
a Communication System**

5                   ***Cross-Reference to Related Applications***

This application is related to the following U.S. applications, all of which are owned by the same assignee as the assignee of this application and all of which are incorporated by reference in their entirety:

10                  Device, System and Method for Spectrally Shaping Transmitted Data Signals, to Vedat Eyuboglu and Pierre Humblet, filed on even date herewith;

                  Device, System and Method for Adaptive Self-Noise Cancellation for Decision-Directed Timing Recovery to Jian  
15 Yang, filed on even date herewith; and

                  System and Device for, and Method of, Detecting, Characterizing, and Mitigating Deterministic Distortion in a Communications Network, to Vedat Eyuboglu and Pierre Humblet, filed on even date herewith.

20

***Background***

1. Field of the Invention

25                  The invention relates generally to communication systems and, more particularly, to high-speed modem communications over a telephone network, possibly having an analog local loop.

## 2. Discussion of Related Art

There is an increasing demand for data communications and, in particular, for communication systems with increasingly higher transmission rates. With the advent of the Internet and multimedia, this demand is not expected to wane any time soon.

To date, communication systems have evolved with attempts to meet these demands. For example, one of the more popular communication paradigms uses modems connected to local loops of a conventional telephone network. These conventional systems are largely desirable because of their leverage of existing telephone network infrastructure. In short, a user needs to make only a relatively small investment for a modem and has to pay relatively modest line charges. To meet user demand for higher transmission rates, modem-communication standards have evolved with each generation including capabilities to support higher transmission rates.

Unfortunately, the rate of growth of the transmission rates of modems has slowed as transmission rates approach the information theoretic limits of the telephone channel. Consequently, users who want higher transmission rates are forced to use alternative communication networks, such as ISDN, in their homes or offices, rather than the conventional public switched telephone network (PSTN) with analog local loops. Though these alternative arrangements provide higher transmission rates, the equipment and line charges are high.

There is, therefore, a need in the art for a device and system for, and method of, communicating at higher

information rates on conventional telephone networks having conventional analog local loops.

### ***Summary***

5       The invention includes a method of, and system and device for, forming an actual alphabet of symbols to be used in a communication system. In this fashion, the communication system may use the actual alphabet to detect symbols rather than use a proscribed alphabet. To do this, the invention  
10       causes the communication system to transmit a predefined sequence of symbols. These symbols are then received and processed to form an actual alphabet of estimate symbols of the transmitted symbol. The actual alphabet may deviate from the proscribed alphabet. Nonetheless, the actual alphabet is  
15       the one used in the communication system, and the forming of an actual alphabet may be exploited to decode the transmitted symbols to prevent known types of erroneous symbol interpretations.

      Among other things, the invention allows information to  
20       be transmitted on a conventional local loop at rates higher than previously considered as the theoretical limits. An exemplary embodiment includes an analog adapter that responds to baseband line interface signals transmitted on to a channel and produces an estimate of the most-probably-  
25       transmitted line interface signal therefrom. The decoder includes a an alphabet of symbols, stored in a storage medium. Each symbol is an estimate of a corresponding line interface signal from the set of line interface signals transmittable on

to the channel. One embodiment uses a subset of the potentially transmittable symbols to form a signal constellation of the baseband line interface signals actually transmitted on the channel. The analog adapter further

5 includes a decoder that cooperates with the alphabet and responds to the received line interface signal to output the estimate of the most-probably-transmitted line interface.

In this fashion, the line interface can produce low fidelity versions of signals specified by a predefined  
10 companding algorithm and the analog adapter will reduce the likelihood of erroneous predictions possible if the alphabet held high fidelity versions of the signals.

An exemplary embodiment includes mechanisms for equalizing sampled versions of the received line interface  
15 signal in which the mechanisms are adaptable to an error signal indicative of the accuracy of the above estimates.

In this fashion, the analog adapter may accurately equalize signals to combat ISI and other forms of noise, while also operating in an environment with low fidelity line  
20 interface signals.

### ***Brief Description of the Drawing***

In the Drawing,

Figure 1 shows a conventional telephone system having  
25 local loops;

Figure 2 is an architectural diagram of an exemplary embodiment of the invention;

Figures 3A-B are architectural diagrams showing a conventional line interface, in part, a local loop, and a decoder of an exemplary embodiment of the invention; and

Figures 4A-4B are architectural diagram of a decoder of  
5 an exemplary embodiment of the invention.

### ***Detailed Description***

This invention allows binary information to be transmitted on conventional telephone networks that include  
10 conventional digital backbones, line interfaces, and analog local loops at transmission rates higher than presently achievable with existing modem standards such as V.34. This is achieved by viewing the conventional network from a new perspective, in which certain sources of "noise" which limit  
15 the achievable bit rate are avoided with new processing techniques in an analog adapter at a user site. The invention improves the system's information capacity and concomitantly achieves higher transmission rates without requiring costly infrastructure, such as ISDN lines or the like, at the user site.

20 To better understand the invention, certain aspects of a conventional telephone network are described. This is done to explain the various sources and forms of "noise" that limit the information capacity of a conventional arrangement and that are addressed with the invention. Afterwards, the  
25 architecture and operation of the invention are described, followed by a description of the invention's mechanisms for combating particular forms of "noise," in particular, nonlinear

distortion by the line interface and intersymbol interference (ISI).

A conventional telephone network 100 is shown in figure 1.

5 What are typically interpreted as analog signals enter and exit the network 100 at "local loops" 140 and 150. Each signal on loop 140 and 150 is received by a corresponding line interface 120 and 130, or local switch, and each line interface communicates with another via a backbone digital network  
10 110.

Under conventional operation, a signal 175 is sent to a first site 170, which emits an analog signal, for example, representative of a voice signal or a binary information, on the local loop 140. The line interface 120 samples and quantizes  
15 the analog signal and outputs an octet 125, representative of the analog signal 140.

More specifically, the analog signal 140 is quantized according to a known set of rules, or a companding algorithm, such as  $\mu$ -law or A-law, which specifies the quantization's  
20 amplitude levels. The  $\mu$ -law and A-law quantization rules involve unequally-spaced quantization steps, i.e., non-linear quantization, that were chosen to map to the inherent characteristics of speech. The quantized signal is then encoded into octets 125.

25 The backbone 110 receives the octet 125 and, though not shown, also receives octets from other sources, such as other line interfaces. Using known techniques, the backbone 110 merges octets from the various sources and transmits and

routes the data to various line interfaces, e.g., 130. Modern backbones transmit data at a rate of 64,000 bits per second (8,000 octets per second). Eventually an octet 125', which is similar but not necessarily identical to the original octet 125, is transmitted to the line interface 130 corresponding to the signal's destination site 160.

The line interface 130 essentially inverse quantizes and further processes the received octet 125' to create on loop 150 an analog signal, which is an "estimate" of originally-transmitted signal 140. Loop signal 150 is called an "estimate" because information may have been lost in the quantization and inverse quantization processes of the line interfaces. Signal 150 is then transmitted to the site 160, where it may be used to recreate a voice signal or a binary sequence.

Information may analogously flow in the opposite direction. Destination 160 provides analog signal on loop 150 to line interface 130. Line interface 130 samples and quantizes the signal on loop 150 to provide a sequence of octets 135 to backbone 110. Backbone 110 routes these octets and provides a similar sequence of octets 135' to line interface 120. Line interface 120 provides analog signal on loop 140 to be received by the site 170.

When used for conventional data communications, as opposed to voice communications, the sites 160 and 170 may each include a modem for modulating and demodulating the analog signals on the local loops 140 and 150. A conventional modem at the site 170, for example, will receive a sequence of

bits 175 from some form of an information source (e.g., a server) and modulate the bits and transmit the modulated signal, according to a communication standard, such as V.34. The modulation technique may use the information contained in  
5 code 175 to alter the amplitude and phase of the signal to be sent on loop 140. The modulated signal is routed to the line interface 120 where it is sampled and quantized, as outlined above. Eventually a representative signal is received by the other modem at site 160, where it may be demodulated and  
10 transmitted to computer 180.

Systems following the above conventional arrangement have achieved transmission rates of approximately 30 Kb/s, the conventionally-accepted view of the telephone channel's capacity. This accepted limit of capacity is dependent on the  
15 "noise" in the system, and in particular, the quantization noise of the line interfaces.

The invention attains higher transmission rates yet operates in arrangements having conventional analog local loops, unlike the ISDN and similar approaches, outlined above.  
20 In short, the invention is able to attain these advantages by considering the conventional network from a new perspective. Under this new paradigm, the invention reconsiders, and where appropriate combats with new processing techniques, the various forms of "noise" that limit the information capacity.

25 More specifically, the invention treats a signal  $s(t)$  on the local loop 150 as a discrete baseband signal and the inverse quantization process in line interface 130 as a baseband modulation that yields the baseband line interface

signal  $s(t)$ . The modulation technique akin to PAM in that a signal's amplitude is modulated but different than PAM in that the amplitudes of adjacent signal points differ non-linearly. The signal  $s(t)$  is in the form

5

$$s(t) = \sum_n a(v_n)g(t - nT) \quad (1)$$

In equation (1),  $v_n$  represents the octets 125' received from the digital backbone network 110;  $a(v_n)$  represents the  
10 transformation of that signal 125' according to the relevant quantization rules, e.g.,  $\mu$ -law;  $T$  equals the sampling interval of the system, e.g., 125  $\mu$ s; and  $g(t)$  is an interpolation function, which is bandlimited to approximately 4000 Hz.

The new perspective yields powerful results the most  
15 important of which is that, unlike conventional systems, embodiments of the invention are not limited in their capacity to carry information by the quantization noise inherent in the line interfaces. An exemplary embodiment attains 56 Kb/s.

To better understand the new paradigm, refer to system  
20 200 shown in figure 2. In system 200, backbone 110, line interface 130, and computer 180 remain unchanged from the conventional components, outlined above. A first site 270, such as an Internet server site, communicates with a digital adapter 220, or digital modem, by sending signals over a high  
25 speed link 240. The digital adapter 220 sends a sequence of octets 225 to backbone 110. Analogously to that described above, backbone 110 sends a similar sequence of octets 225' to conventional line interface 130. Line interface 130 then

inverse quantizes octets 225' and transmits the baseband-modulated, line interface signals, outlined above, on loop 250. Analog adapter 260 receives the baseband signal and may, in turn, possibly equalize and sample the baseband signal, detect  
5 the binary information in the demodulated signal, and sends the results to computer 180. A reverse path from analog adapter 260 to digital adapter 220 may be constructed using conventional modem techniques, for example, using V.34 technology.

10 In an exemplary embodiment, useful for description, signal 240 represents a sequence of bits. These bits are encoded in digital adapter 220 into a sequence of octets 225 which travel to the line interface 130 with minimal alteration. At the line interface 130, the received octets 225' are used to  
15 construct an analog baseband modulated signal on loop 250 according to equation (1) and as specified by the relevant  $\mu$ -law or A-law rules. This latter step, from the perspective of the invention, is now considered as baseband modulation, or a variant of PAM, in which the signal constellation corresponds  
20 to a subset of the quantization levels of the  $\mu$ -law or A-law rules (more below about subset). The analog baseband signal is received by the analog adapter 260, which then samples the received baseband signal at the symbol rate and, possibly after equalization, detects the binary information in the sampled  
25 signal with the results being sent to computer 180, for example. Among other things, the exemplary arrangement 200, unlike the conventional arrangement 100, takes advantage of the fact that there is no analog local loop on one side of the

connection and concomitantly avoids quantization noise as a limiting factor to the system's transmission capacity.

Under the new paradigm of figure 2, the system 200 is theoretically capable of transmitting data at rates of 64,000  
5 b/s and more precisely at the rate of the backbone 110, i.e., 8,000 octets per second. (Consequently, if the backbone operated at a faster rate, the transmission rate of the invention could scale correspondingly) To achieve the 64,000 b/s rate, however, all of the quantization levels must be used  
10 in modulating the baseband signal; that is, each of the quantization levels would correspond to a signal point of a 255 point, one-dimensional constellation. ( $\mu$ -law and A-law have 255 quantization levels)

An exemplary embodiment trades some of the  
15 theoretically possibly bandwidth for noise resistance. In particular, though quantization noise is alleviated, noise resistance may help combat other noise in the telephone channel.

More specifically, the spacing between some of the  
20 adjacent quantizer levels in  $\mu$ -law and A-law is relatively small. Consequently, the "minimum distance," or  $d_{\min}$ , is small of a signal constellation that includes these small  $\mu$ -law and A-law levels as signal points. ( $d_{\min}$  is a known parameter for characterizing the performance of a signal constellation in an  
25 uncoded system, and in short,  $d_{\min}$  refers to the shortest "distance" between different levels in a signal constellation. The distance may be measured according to different known metrics, such as Euclidean distance or Hamming distance.)

An exemplary embodiment of the invention, exploits the inherent non-linear characteristics of the  $\mu$ -law or A-law rules to achieve an acceptable  $d_{\min}$  yet retain substantially improved transmission rates. The above exploitation may be best illustrated by comparing a uniformly spaced (PAM) signal constellation, with a non-uniform  $\mu$ -law or A-law signal constellation. To double  $d_{\min}$  of a uniformly spaced (PAM) signal constellation or the non-uniform  $\mu$ -law or A-law signal constellation, the amplitude difference between the closest signal points needs to be doubled. To do this for a uniformly spaced (PAM) signal constellation, requires that every other level would need to be eliminated. To do this for non-uniform  $\mu$ -law or A-law signal constellation, a significantly smaller percentage of levels needs to be eliminated. This is so, because the spacing between adjacent levels grows non-linearly and rapidly. Much less than half of the levels need to be eliminated before the smallest spacing between remaining levels is doubled.

To this end, the above embodiment attains an advantageous trade-off of transmission rate for noise resistance by using a signal constellation that excludes some of the quantization levels of the line interface 130. That is, the alphabet used by the system 200 will exclude some octets 225 that would be otherwise inverse quantized to levels that would result in small spacings relative to other symbols in the alphabet.

Consequently, a preferred embodiment uses a subset of the  $\mu$ -law or A-law quantization levels as valid levels in the

signal constellation. Using a subset allows the system to attain transmission rates approaching 56 Kb/s, yet attain desirable levels of noise resistance.

The above system 200 and corresponding paradigm  
5 departs from the conventional arrangement 100 to attain significant advantages, but it also creates design problems and issues with no parallel in the conventional arrangement. Among other things, the new arrangement creates problems of

- 10 1. ensuring that the signal 225 is appropriately modified, or spectrally shaped, to improve overall performance;
2. ensuring that the analog adapter 260 has precise enough timing to properly sample the baseband modulated signals received on loop 250;
- 15 3. combating certain distortion introduced by the digital backbone network such as "robbed bit signaling," which otherwise would effectively act as a form of noise limiting the system's capacity;
- 20 4. handling intersymbol interference (ISI) generated by the line interface 130 and the loop 250 so that the transmitted binary information sent by the source 270 may be recovered; and
- 25 5. combating various forms of system-introduced noise, such as memory-less nonlinear distortion from the line interface 130, so that the binary information transmitted sent by the source 270 may be recovered.

### I. The Digital Adapter and Spectral Shaping

The digital adapter 220 receives data from the site 270, for example, in the form of a bit stream from a Local Area  
5 Network (LAN) or the Internet. The digital adapter 220 encodes the incoming bit stream 240 into a sequence of octets 225, which are transmitted to the backbone 110.

The line interface 130 converts the sequence of received octets 225' into a sequence of quantization levels. In certain  
10 situations, it is desirable to shape the frequency spectrum of this sequence to combat the effects of certain forms of distortion. For example, it may be desirable to avoid placing any energy at DC to avoid certain distortion that may be created by such energy. Although such distortion may be  
15 relatively tolerable for voice communications, it may present a significant impairment to data communications.

The system uses a novel mechanism to spectrally shape the sequence of quantization levels to be transmitted. The spectral shaping assures that the data attain the desired  
20 characteristics, while minimizing the shaping's impact on achievable transmission rates. This aspect is described in the U.S. Pat. Apl. entitled Device, System and Method for Spectrally Shaping Transmitted Data Signals, identified and incorporated above.

25

### II. The Analog Adapter and Timing Recovery to Properly Sample Signals

Referring briefly to figure 4B, the analog adapter 260 includes a section 440 for sampling the baseband signal received from the local loop 250 possibly after equalization and a section 450 for detecting, or estimating, the binary information in the demodulated signal 445.

The system includes novel mechanisms for providing the timing signals used for sampling the signals 250. This aspect is described in the U.S. Pat. Apl. entitled Device, System and Method for Adaptive Self-Noise Cancellation for Decision-Directed Timing Recovery, identified and incorporated above.

### III. Combating Robbed-Bit Signaling

Robbed bit signaling is a technique used in the telephone network to accomplish various signaling functions. Robbed bit signaling can modify the octets as they are being transmitted across the digital network. In this regard, robbed bit signaling is a form of distortion or noise that can limit the capacity of the system.

The description above alluded to this aspect when stating that the sequence of octets 225 entering the backbone 110 is not necessarily the same as the sequence of octets 225' exiting the backbone 110. The two sequences may differ depending upon the presence and type of robbed bit signaling.

The system includes novel mechanisms for handling robbed bit signaling. This aspect is described in the U.S. Pat. Apl. entitled System and Device for, and Method of, Detecting,

Characterizing, and Mitigating Deterministic Distortion in a Communications System, identified and incorporated above.

#### IV. Controlling Inter-Symbol Interference (ISI)

5       The backbone 110 and line interface 130 were designed and constructed for voice communications. One consequence of the design is that an interpolation filter (420, see fig. 4A) typically found in the line interface 130 does not satisfy Nyquist's criterion when signaling at 8000 baud, causing ISI on  
10   the signal received by the analog adapter 260.

      To handle ISI, the invention uses a novel arrangement of an equalizer and a level decoder. Because the inventive arrangement for controlling ISI is also used to combat system introduced noise, to avoid a redundant description, the  
15   arrangement is discussed in the next section only.

#### V. Combating System-Introduced Noise

      To better understand the invention's novel mechanisms for combating ISI and other noise, refer to figures 4A-B. In  
20   figure 4A, only the parts of the line interface 130 and loop 250 that are material to understanding the invention are shown. In figure 4B, the analog adapter 260 is shown as a high-level architectural diagram.

      The line interface 130 includes a digital-to-analog  
25   converter (D/A converter) 410 and a low pass filter (LPF) 420, or interpolation filter. The D/A converter 410 is responsible for converting the received sequence of octets 225' into a sequence of quantization levels as outlined above. That is, the

D/A converter 410 will receive an octet 225',  $v_n$ , and construct a signal 415,  $a(v_n)$ , having an amplitude level corresponding to the octet 225' and the relevant  $\mu$ -law or A-law rules (more below). The resulting sequence of levels 415 is then sent to  
5 LPF 420, which shapes the sequence and sends the resulting line interface signals 416 on to the channel 430. For descriptive purposes, the channel 430 may be modeled as having an impulse response  $g(t)$ . Thus, the signal exiting the channel at this point is modeled as  $s(t)$ , described above in  
10 equation (1). Signal  $s(t)$  is subject to the addition 435 of a noise component  $n(t)$ , yielding the analog signal received on loop 250 by the analog adapter 260.

The analog adapter 260, shown in figure 4B, includes a section 440 that is responsible for sampling the signal 250.  
15 The various components 441-443, responsible for timing recovery, are described in the related applications that were identified and incorporated above and will not be described here. Suffice it to say that demodulated signal 445 is a sampled version of loop signal 250.

20 The analog adapter 260 also includes an equalization and detection section 450 that is responsible for compensating for the linear distortion and then "interpreting" the resulting sampled equalized sequence 465,  $r_n$ . In this regard, "interpretation" means analyzing the sequence 465 to detect  
25 which sequence 225' of octets were sent. Since this sequence is nearly identical to the originally-transmitted sequence of octets 225, the original transmitted binary information can be recovered. (The invention's handling of robbed bit signaling

compensates for any discrepancies between 225 and 225'.)  
Among other things, this detection must account for the noise  
 $n(t)$  on the channel, the effects of the channel  $g(t)$ , the  
presence of ISI, and the effects of non-linearities in the D/A  
5 converter 410.

Beginning with the latter, the invention assumes that  
real-world systems will not precisely follow the  $\mu$ -law or A-  
Law quantization levels proscribed in ITU Recommendation  
G.711. Instead, the invention assumes that the line interface  
10 130 will be low fidelity with regard to the accuracy of the  
transmitted levels  $a(v_n)$  vis-à-vis the proscribed levels.

In other words, the line interface 130 will produce, in  
response to a given octet 225', a level not having the precise  
amplitude specified by the  $\mu$ -law or A-law rules as  
15 corresponding to the given octet. Instead, the invention  
assumes that the line interface 130 will produce a level having  
an amplitude level that varies from the specified amplitude  
and, moreover, that the amount of variation between the real  
amplitude and the specified amplitude will depend on the  
20 specified amplitude level. For convenience, using  $y(i)$  to  
designate a level taken from the A-law or  $\mu$ -law  
specifications, representing the  $i$ 'th level in the signal  
constellation where  $i$  is an integer between 0 and  $M-1$ , the  
actual level 415 produced by D/A converter 415 may be  
25 mathematically described as follows:

$$x(i) = y(i) + \Delta(y(i)) \quad (2)$$

In equation (2), the error component  $\Delta(y(i))$  describes how much the actual level  $x(i)$  varies from the level  $y(i)$  specified in A-law or  $\mu$ -law. Moreover, although the error component is  
5 described as a function of the particular specified level, it should be appreciated that the error component is not known *a priori* and that the underlying relationship, defining the function  $\Delta(y(i))$ , may change slowly over time and is likely to change from one connection on the telephone network to the  
10 next.

Without more, the error component is a source of non-linear noise that could affect the system's information-carrying capacity. Unless corrective steps are taken, the low fidelity line interface 130 could cause incorrect predictions,  
15 or interpretations, of the line interface signal 416 that was transmitted on to the channel 430.

The problem of possible erroneous prediction is illustrated with figures 5A-B. Figure 5A shows two points  $y(1)$  and  $y(2)$  corresponding to two of the "signal points" of the  
20 one-dimensional constellation of an exemplary embodiment. Each signal point corresponds to an amplitude level and may be considered as an information-carrying "symbol." The set of symbols may be considered as an "alphabet." Graphing all symbols of an alphabet constructs a "constellation,"  
25 representative of the code.

Assuming that the alphabet mirrors the  $\mu$ -law or A-law specified amplitude levels,  $y(1)$  corresponds to one of the 255

specified amplitude and  $y(2)$  corresponds to another, possibly adjacent specified amplitude level. In a noise-free, distortion-free system, the signals 250 received possibly equalized and sampled by the analog adapter 260 will only have the amplitude levels  $y(1)$ ,  $y(2)$  and so on.

As alluded to above, however, the signal 250 includes additive noise  $n(t)$ , non-linear distortion due to errors in the D/A converter, ISI, and the like. Thus, the eventually received and sampled sequence  $r_n$  should not be expected to fall right on a signal point of the constellation.

In a conventional system, a decoder will analyze the received sequence  $r_n$ , for example, using distance metrics, to detect the sequence of levels that most probably was sent that would have yielded the received signal. For example, a conventional decoder, responsible for detecting the transmitted signals of the simple, memory-less signal constellation of figure 5A, will predict that the transmitted signal was  $y(1)$ , because the distance  $d[r_n, y(1)]$  between  $r_n$  and  $y(1)$  is smaller than the distance  $d[r_n, y(2)]$  between  $r_n$  and  $y(2)$ . The smaller distance  $d[r_n, y(1)]$  is interpreted as meaning that it was more likely that  $y(1)$  was transmitted. When coding is employed, e.g., trellis codes, analysis of metrics would involve comparing distances between sequences of levels rather than individual levels.

Figure 5B illustrates how the conventional arrangement may yield incorrect predictions when the characteristics of a low fidelity line interface 130 are considered. Figure 5B includes all of the items of figure 5A and also illustrates two

new items  $x(1)$  and  $x(2)$ .  $x(1)$  and  $x(2)$  represent the actual levels of the transmitted signal point 416, i.e., the signals described by equation (2). Thus,  $x(1)$  and  $x(2)$  represent the true levels the line interface 130 actually uses. In other words, when interface 130 means  $y(1)$  it actually sends  $x(1)$  and when it means  $y(2)$  it sends  $x(2)$ .

As shown, if a decoder operates on the ideal world alphabet, specified with  $y(1)$  and  $y(2)$  as a partial alphabet, it will predict that  $y(1)$  was transmitted, because the distance  $d[r_n, y(1)]$  is smaller than the distance  $d[r_n, y(2)]$ . This determination would be incorrect, however. As shown, appreciating the actual distortion and its effect on the real-world alphabet, specified with  $x(1)$  and  $x(2)$  as a partial alphabet, it is more likely that the transmitter sent symbol  $x(2)$  because the distance  $d[r_n, x(2)]$  between the received sequence  $r_n$  and the actual signal point  $x(2)$  is smaller than the distance  $d[r_n, x(1)]$ .

To address this problem of error, an exemplary embodiment of the invention estimates the actual alphabet, i.e., the actual signals transmitted, and then uses those estimates in an improved level decoder to predict the most-probably-transmitted signal. In this fashion, the exemplary decoder will avoid the erroneous interpretations illustrated in figure 5B. As will be explained below, the exemplary decoder computes metrics with respect to the estimate of the actual signal constellation used by the line interface, rather than the pre-specified A-law or  $\mu$ -law quantization levels specified in Recommendation G.711.

Another aspect of the invention is to estimate the actual alphabet in the presence of other linear distortion. An exemplary embodiment of the invention compensates for such distortion with an equalizer, e.g., a linear equalizer or a decision-feedback equalizer. On severely distorted channels, a linear equalizer may be combined with a maximum-likelihood sequence decoder (MLSD) or a suboptimum version thereof, in which the linear equalizer removes part of the ISI and the Viterbi equalizer removes the rest.

First consider the somewhat idealized case in which there is no linear distortion in the system, or the distortion is already equalized by some other means. In such a system, the received sequence  $r_n$  will be of the form:

$$r_n = x_n + N_n, \quad (3)$$

where  $x_n$  is the actually-transmitted symbol, also represented by the index  $i_n$ . The decoder 470 keeps in storage an alphabet  $dX_n = \{dx(0), dx(1), \dots, dx(M)\}$  of symbols which represent the decoder's estimate of the actual alphabet. The decoder includes logic to compute metrics based on the received symbol  $r_n$  and symbols  $dx_n = dx(i_n)$  from the estimated alphabet.

An exemplary embodiment includes logic to generate an estimate of the actual alphabet, as follows. During a training period the digital adapter 220 sends a known sequence of octets 225, corresponding to signal point indices  $i_n$ . The decoder 470 eventually receives a corresponding baseband signal on loop 250 and maintains an ordinary average of the

received level. When performed over many symbols, the averages will accurately represent the values of the actual levels.

Such averages can also be determined iteratively using the Least-Mean Square (LMS) algorithm. In this case, the decoder would compute the error  $e_n$ , between the received signal  $r_n$  and the estimate  $dx(i_n)$  of the actual signal  $x(i_n)$  taken from the present alphabet, and will then update this estimate according to:

10

$$dx(i_n) \rightarrow dx(i_n) + \alpha e_n, \quad (4)$$

where  $\alpha$  is a small update coefficient. The value of  $\alpha$  depends on the signal constellation, but is typically a small fraction of the average magnitude of the actual signal points.

When the transmitted index sequence  $i_n$  is unknown in the receiver, as it would normally be during data transmission, the decoder can compute the error signal as the difference between the received signal  $r_n$  and the decoders estimated level  $dx(i_n)$  from the present alphabet 471. That value of that level is then updated according to the averaging scheme or the LMS update formula given above in equation (4).

When the above scheme is used with an adaptive linear equalizer, a new problem arises. When the equalizer coefficients 461 and the decoder alphabet 471 are updated jointly, both the estimates in the alphabet and the equalizer coefficients may converge towards zero. An exemplary

embodiment of the invention avoids this undesirable operation by constraining the sum of the magnitudes of the equalizer coefficient to be equal to some non-zero value.

As shown in figure 4A, the equalizer 460 outputs the  
5 sequence  $r_n$  465 which is equalized using a transversal filter arrangement with tap coefficients  $w$ . The difference between the equalized signal, i.e., one that is processed to control ISI and the like, and the estimate  $dx(i_n)$  of the actual level that was most probably sent to yield that equalized signal may be  
10 expressed with the following equation:

$$e = r - dx(i_n) \quad (5)$$

As mentioned above, one known solution to minimize the  
15 energy of the error described in equation (5) would force  $dx(i_n)$  and tap coefficients  $w$  to zero. Though this solution minimizes the average error, it also removes all information and must, therefore, be guarded against.

To protect against the uninformative solution, an  
20 exemplary embodiment imposes the constraint that the tap coefficients  $w$  have a *norm* equal to 1. That is,

$$\| w \| = 1 \quad (6)$$

25 (The mathematical concept of a norm is known).

To update the decoder's estimate of the alphabet 471, the decoder could use the same iterative LMS algorithm described

earlier with equation (4). In this fashion, the decoder's 470 alphabet 471 is updated iteratively. The initial alphabet could correspond to the nominal values proscribed by the ITU Recommendation and the initial update may take part from a training sequence included in an initialization procedure. For example, all symbols in the alphabet may be sent many times to make sure that the system is sufficiently exercised to create an initial real-world alphabet 471 in decoder 470.

Analogously to the above, the adaptive equalization of signals is best accomplished when performed using the symbols from the estimated alphabet as a reference to form the error signal  $e_n$ . That is, the quality of the equalization is judged by comparing the equalized signal to an estimate of the signals actually transmitted rather than by comparing the equalized signal to some pre-defined ideal value. Conventionally, equalization would compare the output of the equalizer with a corresponding expected value of an output, such as a prescribed value of the alphabet. As stated above, the invention dynamically builds an alphabet 471 corresponding to the symbols actually used by the line interface 130, rather than using the pre-specified amplitude levels of the  $\mu$ -law rules, for example.

Thus, an exemplary embodiment updates the  $i$ 'th equalizer coefficient according to the following iterative procedure using known techniques.

$$w_i(n+1) = w_i(n) + \epsilon \cdot x_{i,n} e_n. \quad (7)$$

Here,  $x_{i,n}$  represents the equalizer input signal at the  $i$ 'th coefficient, and  $e_n$  is the error signal described earlier.  $\epsilon$  will be empirically determined, and its value is typically much smaller than the average energy of the equalizer input signal. To prevent the equalizer coefficients from converging towards zero, the coefficients need to be scaled up. This can be accomplished by multiplying all coefficients once in a while by a scale factor  $\beta$ , a number slightly greater than 1.0, or by multiplying  $w_i(n)$  by a similar scale factor  $\beta$ .

The above-described use of creating an error signal by comparing the equalized signal with a corresponding symbol from a dynamically-built estimate of the alphabet may also be used as an error signal when the linear equalizer is followed by a Viterbi decoder (for example, in partial-response systems), or when a decision-feedback equalizer is used. The only difference is that in these cases, the reference signal used in computing the error signal may depend on more than one symbol.

The equalization and detection algorithms can be implemented on conventional DSP hardware or on PC processors using well-known programming techniques.

A prior embodiment of the above invention was described in French Application No. 95-12672 on Oct. 23, 1995 to Pierre Humblet and is incorporated by reference in its entirety.

Although the invention was described in the context of a particular preferred embodiment, namely one that uses the D/A converter in line interface card, it has much broader

applicability. In fact, the invention can be used in any digital communication system, where there the generation of the symbols in the transmitter introduces a non-linearity which causes the transmitted symbols to deviate from their pre-  
5 specified values. The invention also applies to situations where the transmission medium further introduces distortion, thus requiring an equalizer in the receiver.

The present invention may be embodied in other specific forms  
10 without departing from the spirit or essential characteristics. The described embodiments are to be considered in all respects only as illustrative and not restrictive.

What is claimed is:

1. A method of forming an actual alphabet of symbols to be used in a communication system so that the communication system may use the actual alphabet to detect symbols rather than use a proscribed alphabet, the method comprising the steps of:
  - causing the communication system to transmit a predefined sequence of symbols;
  - receiving and processing the transmitted symbols to form an actual alphabet of estimate symbols of the transmitted symbol, wherein the actual alphabet may deviate from the proscribed alphabet.
2. The method of claim 1 wherein the communication system includes a low fidelity line interface that inherently transmits signals that deviate from signal levels proscribed by a predefined companding algorithm, and
  - wherein the causing step includes the substep of sending a sequence of octets to the line interface, the octet sequence being known to cause the line interface to transmit a sequence of signals that are a subset of the signals transmittable by the line interface, and wherein the subset defines a signal constellation of the communication system.

3. The method of claim 2 further comprising the step of creating an initial alphabet having nominal symbols defined by the proscribed alphabet, and

5                wherein the receiving and processing step includes the substep of updating the stored alphabet by averaging a stored symbol with the received symbol according to a defined averaging function, and

                 wherein the actual alphabet is formed after  
10                repeating a predefined number of times the step of sending the octet sequence to the line interface.

4. The method of claim 3 wherein the receiving and processing step includes the step of equalizing the signal  
15                transmitted by the line interface to form the received symbol used in the updating step.

5. The method of claim 4 further comprising the step of forming an error signal as the difference between the received  
20                symbol and the estimate symbol corresponding to the received symbol and wherein the defined averaging function uses the error signal as an averaging function variable in the updating step.

25                6. The method of claim 5 further comprising the step of adapting the equalizing signal according to a predefined update function that uses the error signal as an updating function variable.

7. In a communication system that transmits on to a channel signals from an actual signal constellation, in which the actual signal constellation may deviate from a proscribed signal constellation, an adapter comprising:

5 a storage medium for holding an alphabet having estimate symbols of a corresponding signal transmitted on to the channel; and  
a decoder, responsive to a received version of the transmitted signal, the decoder cooperating with the  
10 alphabet and having a decoder output, selected from the alphabet of estimate symbols, and carrying an estimate of a most-probably-transmitted signal.

8. The adapter of claim 7 further comprising  
15 an error signal generator, responsive to the received version of the transmitted signal and to the decoder output, having an error output indicative of the accuracy of the estimate carried on the decoder output; and  
wherein the decoder includes logic for updating the  
20 alphabet in response to the error output.

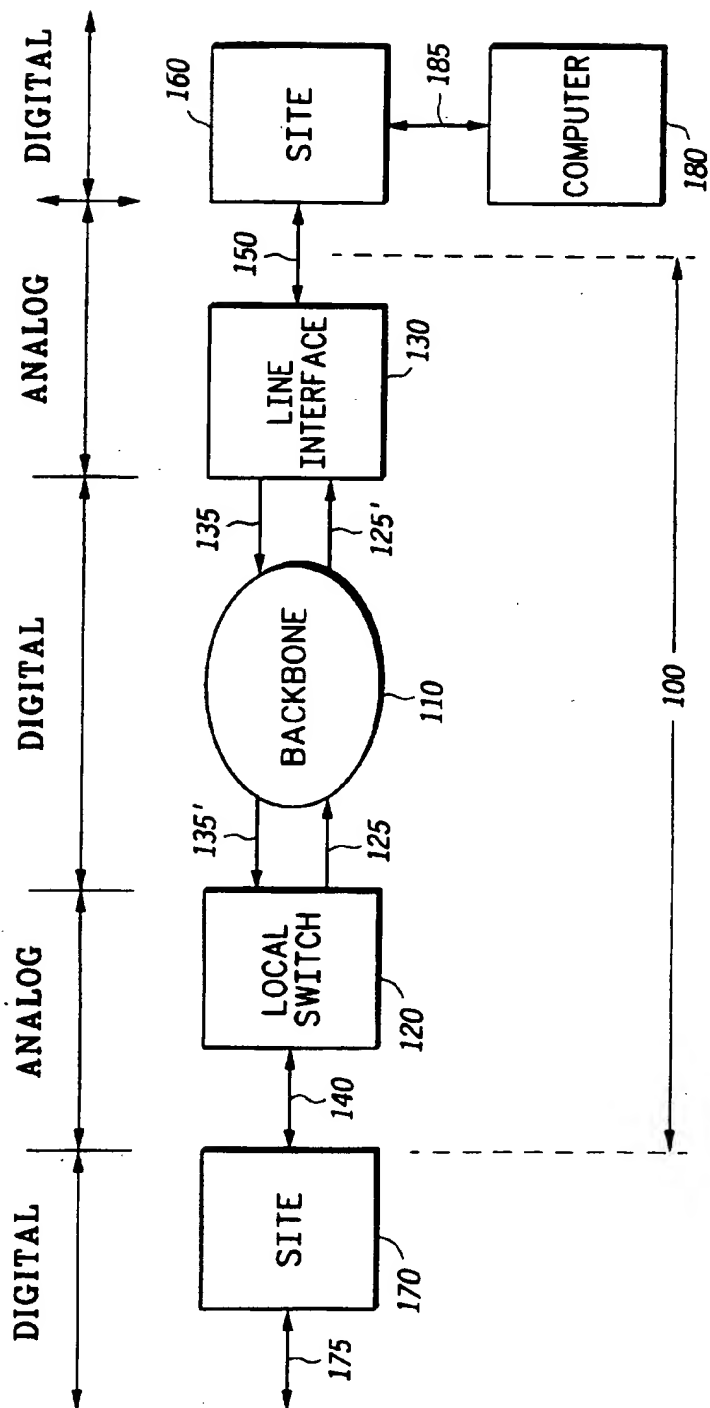
9. The adapter of claim 8 further comprising  
an equalizer, responsive to the received version of  
the transmitted signal, having an output carrying an  
equalized version thereof, and  
5 wherein the equalized version is input to the  
decoder and the error signal generator and wherein the  
error output is the difference between the equalized  
version and the decoder output.
- 10 10. The adapter of claim 9 wherein the equalizer has  
adaptable tap coefficients that adaptably respond to the error  
output.
11. The communication system and adapter of claim 7  
15 wherein the communication system includes a line interface  
that produces baseband signals on a local loop,  
wherein the amplitude of the baseband signals  
corresponds to but deviates from a proscribed  
companding algorithm,  
20 wherein the adapter is couplable to the local loop,  
wherein the communication system includes means  
for causing the line interface to transmit on to the  
channel signals defining an actual signal constellation of  
the line interface which deviates from the ideal signal  
25 constellation proscribed by the companding algorithm,  
and

wherein the adapter includes initialization means, cooperating with the means for causing, for forming the alphabet of estimates of the transmitted signals.

5 12. The communication system and adapter of claim 11 wherein the companding algorithm is defined by the  $\mu$ -law rules.

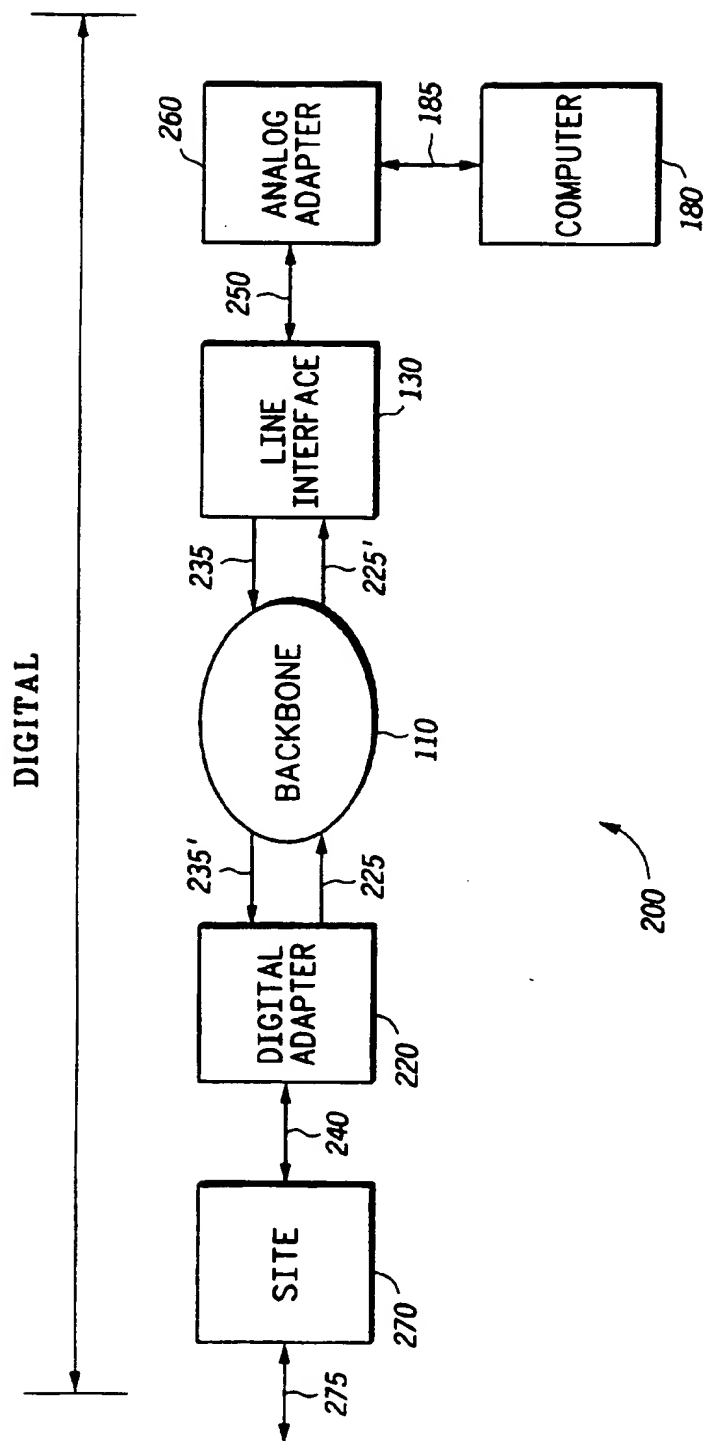
10 13. The adapter of claim 8 wherein the logic for updating the alphabet updates entries  $\hat{a} (v_n)$  of the alphabet with a stored version of entry  $\hat{a} (v_n)$  plus a scaled version of the error output so as to average the entries to form the estimates.

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**FIG. 1**  
-PRIOR ART-

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*FIG. 2*

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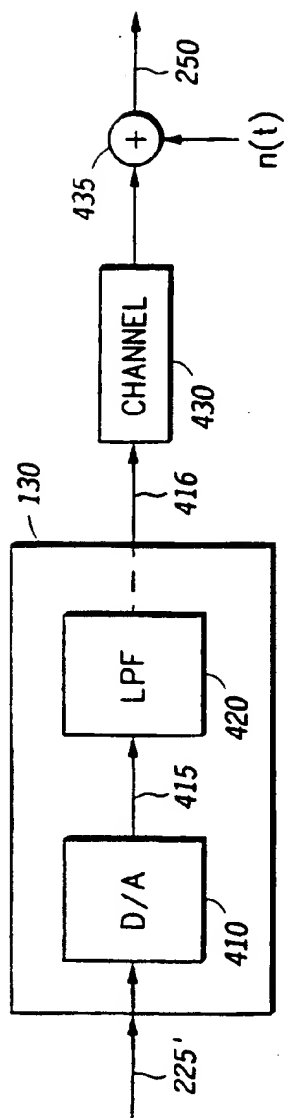


FIG. 3A

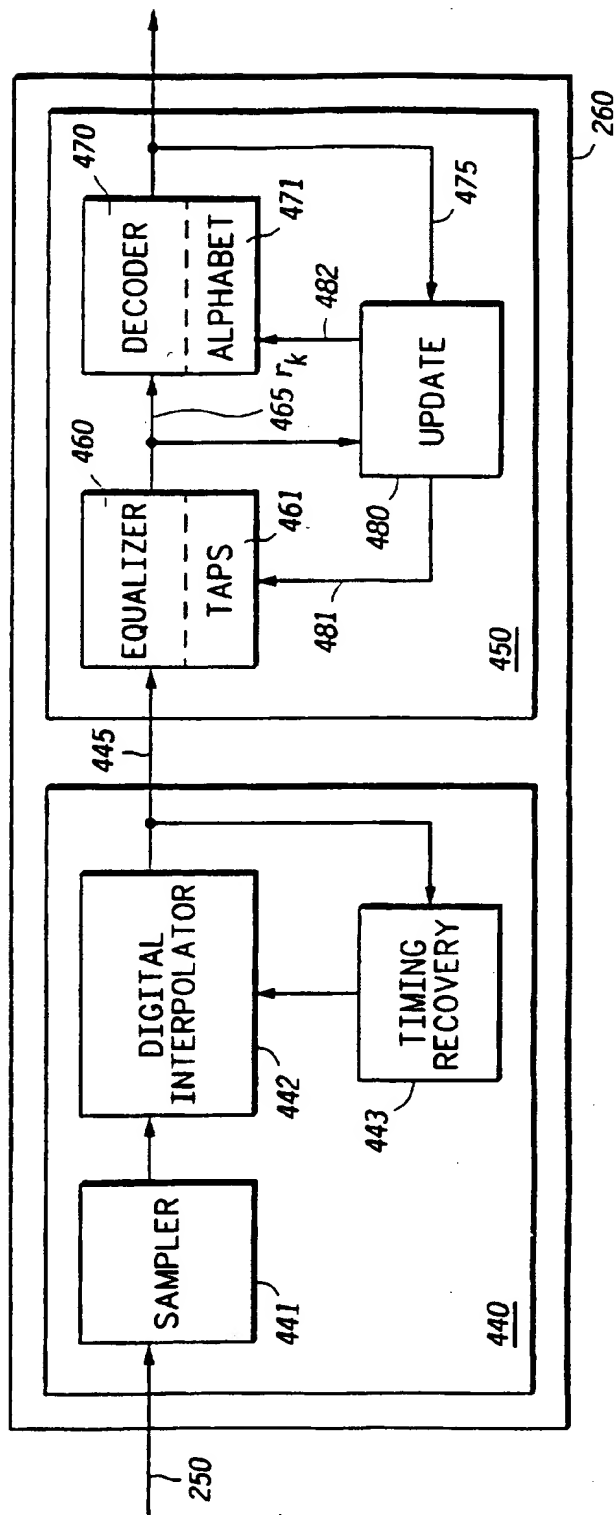
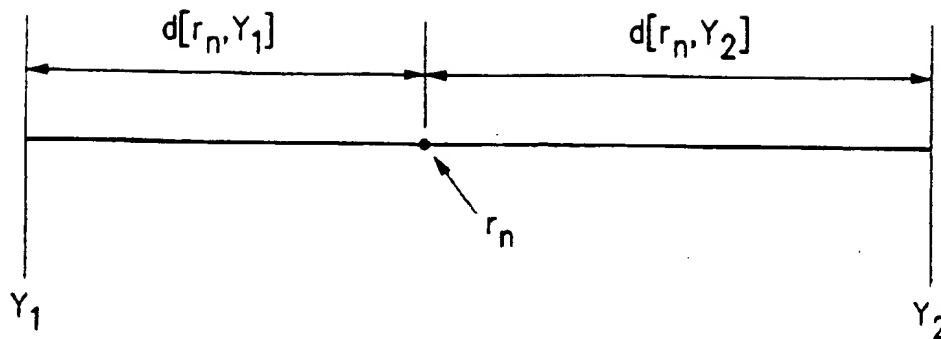
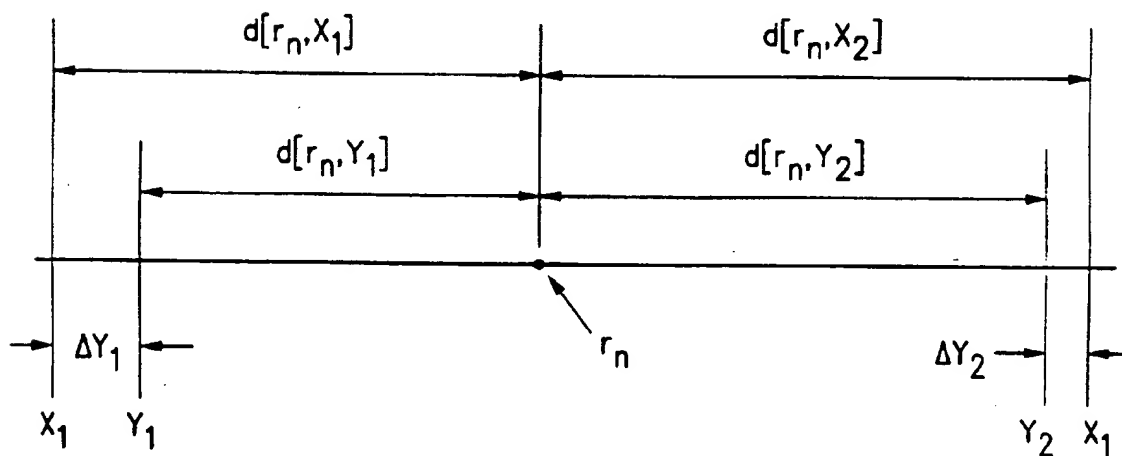


FIG. 3B

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$$d[r_n, Y_1] < d[r_n, Y_2]$$

**FIG. 4A**

$$d[r_n, Y_1] < d[r_n, Y_2]$$

$$d[r_n, X_1] > d[r_n, X_2]$$

**FIG. 4B**

## INTERNATIONAL SEARCH REPORT

International application No.  
PCT/US97/16909

**A. CLASSIFICATION OF SUBJECT MATTER**

IPC(6) : H04L 27/00

US CL : 375/231, 285, 346

According to International Patent Classification (IPC) or to both national classification and IPC

**B. FIELDS SEARCHED**

Minimum documentation searched (classification system followed by classification symbols)

U.S. : 375/222, 231, 232, 284, 285, 286, 287, 324, 340, 341, 346; 371 40.1, 40.2, 41; 341/94, 106, 107; 364/724.19, 724.20

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched  
NONE

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

APS: SYMBOL; COMPAND?; ALPHABET; inventors names

**C. DOCUMENTS CONSIDERED TO BE RELEVANT**

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	WO 96/18261, 13 June 1996, bottom of pg 34 to top of pg 35, abstract, pg 21 last line, page 22 second paragraph 4th line and 6th line noting item 272, Fig 10 with corresponding disclosure.	1-13
Y,P	US 5,659,579 (HERZBERG) 19 August 1997, abstract, Fig's 9 and 13, col 2 lines 6-26, col 8 line 28.	2-6 & 8-13
Y	US 5,040,191 (FORNEY, JR et al) 13 August 1991, abstract, Fig 18 item 99, col 4 lines 6, 52, 54-55, 56-59, col 5 lines 2-3, 12.	1-13
Y	US 5,528,625 (AYANOGLU et al) 18 June 1996, abstract.	1-13

☒ Further documents are listed in the continuation of Box C. ☐ See patent family annex.

* Special categories of cited documents:	*T* later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
*A* document defining the general state of the art which is not considered to be of particular relevance	*X* document of particular relevance; the claimed invention cannot be considered novel or cannot be considered to involve an inventive step when the document is taken alone
*B* earlier document published on or after the international filing date	*Y* document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
*L* document which may throw doubts on priority claim(s) or which is cited to establish the publication date of another citation or other special reason (as specified)	*A* document member of the same patent family
*O* document referring to an oral disclosure, use, exhibition or other means	
*P* document published prior to the international filing date but later than the priority date claimed	

Date of the actual completion of the international search

21 NOVEMBER 1997

Date of mailing of the international search report

12 JAN 1998

Name and mailing address of the ISA/US  
Commissioner of Patents and Trademarks  
Box PCT  
Washington, D.C. 20231

Facsimile No. (703) 305-3230

Authorized officer

WILLIAM LUTHER

Telephone No. (703) 308-6609

## INTERNATIONAL SEARCH REPORT

International application No.

PCT/US97/16909

## C (Continuation). DOCUMENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 5,394,437 (AYANOGLU et al) 28 FEB 95, abstract.	1-13
Y	ROCKWELL INTERNATIONAL, 56 Kbps Communications Across the PSTN, 1996, Internet URL, <a href="http://www.nb.rockwell.com/nr/modemsys/">http://www.nb.rockwell.com/nr/modemsys/</a> . See the section titled 'THE COMMUNICATIONS PATH' 1st paragraph. See section titled 'DEALING WITH THE COMMUNICATIONS PATH' 3rd paragraph.	1, 7
Y	KALET, et al., The Capacity of PCM Voiceband Channels, 1993, IEEE 0-7803-0950-2/93, pages 507-511. See abstract and Figure 1.	1, 7